A Digital Audio System For CATV Applications

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ABSTRACT

With the recent interest in improving broadcast audio quality for the consumer, various systems are being proposed as solutions. After a review of current activities, a description of a digital audio system which offers quality equivalent to the Compact Disc is followed by a discussion of field trial plans.

INTRODUCTION

Audio quality as available to the consumer is improving from both the source and home reproduction perspectives. There are also several approaches being proposed for making similar improvements in the broadcast techniques used to deliver audio to the home, using either direct over the air, satellite, MDS or CATV systems.

The following table illustrates the relative performance of common consumer audio products:

System	Audio S/N dB
Compact Disc	90
HI-FI VCR	80
Cassette (Dolby C)	75
Turntable (LP)	70
Off Air FM Stereo	65
Cassette (Dolby B)	65
MTS (DBX NR)	65
Cassette	57
Cable FM Stereo(-15dB)	55
VCR	47

Many of the above systems suffer from performance which does not achieve the given specifications. Wherever alignment of the system is required, the operator or the consumer is unlikely to provide maintenance to preserve the maximum quality. Also, records and tapes suffer degradation from repeat usage. In general, a specification of 70 dB S/N is considered acceptable with 80 dB S/N considered excellent. As such, only the Compact Disc and HI-FI VCRs are currently considered to fall into the excellent category.

Program Source Improvements

Program source improvements are moving quickly on several fronts. In the recording studio, digital multi-track recorders (such as the Sony PCM 3324) are replacing the current analog based machines. The TV broadcast community is adopting the BTSC MTS stereo standard, CATV satellite delivered services are upgrading with digital and high quality analog transmission, and the Compact Disc is having a major impact on the studio operations of many quality conscious FM broadcasters.

In the home, the Compact Disc system is also taking off very quickly. At present, the acceptance rate is growing at least twice as fast as did VCRs. In addition, users of the system are buying double the average number of discs as they previously purchased as albums. this can be explained by the low cost of home players (under $300 units are available) and the initial need to stock a library with new discs. The net result is that the home listener is moving up in his expectation of audio performance.

Recording Improvements

Recording equipment improvements are also moving into the home at affordable prices. The Beta and VHS HI-FI cassette tape recorders are the main example of this trend. However, there are other aspects such as PCM adapters for older VCRs and Dolby C modes on standard audio cassette recorders. There are strong indications that a new home audio recording system will be unveiled next year by several Japanese manufacturers. This system would provide two hours of record time using a digital audio cassette format.

Transmission System Improvements

Transmission system improvements are following closely on the heels of the source and recording upgrades. Over the air broadcasters have adopted a new stereo standard for TV, the MTS (Multi Channel Sound) system which includes DBX noise reduction. FM broadcasters are mounting a campaign to take full advantage of the quality possible in their current system. Satellite delivered signals for CATV headend use are adopting both digital (Wegener ADM Dolby, M/A-COM VideoCipher II) and analog (Wegener Panda II, Studioline) techniques.

For delivery directly to the home several new alternatives are under development (see references 1,2,3 and 4). CATV equipment vendors are developing both in-band and out-of-band systems. An in-band system delivers audio within the video channel. Examples of this are the Oak Sigma series and M/A-COM VideoCipher II, which use digital technologies, replacing horizontal sync with digital audio. In an out-of-band system, the audio information is broadcast separately in another part of the spectrum. Both analog and digital designs are being suggested. For analog approaches, one alternative is carriage in the FM band in the standard format. A tracking adapter tunes to the proper FM channel when the video channel is changed. Examples of this product are being shown by Pioneer (reference 5) and Westinghouse/Sanyo. A "digital" quality analog system is being shown by Studioline, also of the tracking type. In the digital realm, products have been shown by Sony, Panasonic and Toshiba. Other vendors are known to have both types of CATV systems under development.

Other alternatives include DBS and MMDS (multi channel MDS) delivery systems. In the new DBS products there are digital audio channels being designed in. The MMDS market, which is now being launched, is expected to use the lure of digital audio as one of its selling points. An interesting option in MMDS is to devote a full video channel to audio only. This would provide at least 8 high quality stereo channels.

Why Consider Digital?

Why consider digital technology for the transmission of audio in CATV applications? Certainly, there is a trend towards digital recording and Compact Disc sales to the consumer. Therefore, it will be easy to convince CATV customers that they are getting the best sound when it is digitally broadcast into the home. But as engineers, picking systems which must be compatible with our cable plants, and which must make efficient use of the limited spectrum available, a closer examination of the alternatives is in order. Given these concerns, in-band solutions are attractive, however unless a complete changeout of equipment is possible this is not practical. Therefore, out-of-band solutions are likely to be chosen.

The next table shows the stereo channel efficiency for the major out-of-band contenders, all of which have high quality performance:

System	Channels/6 MHz
M/A-COM PCM digital	8
Sony PCM digital	8
Toshiba PCM digital*	8
Panasonic PCM digital*	12
Dolby ADM digital	16
Studioline analog	20

*These systems also have less efficient modes with S/N 90dB

The PCM (Pulse Code Modulation) systems use various forms of companding and QPSK modulation for greater efficiency. A Dolby ADM (Adaptive Delta Modulation) system designed with integrated circuits available this year from Signetics, is also likely to use QPSK modulation. Studioline, currently the sole analog system for consideration, is based on the Telefunken High-Com companding process and discrete L/R FM modulated channels.

Given the example of Studioline, it is shown that there is certainly not an advantage in terms of bandwidth considerations where digital is concerned. However, this is not to say that there is an inherent disadvantage with digital as improved systems such as the Dolby ADM are perfected. In fact, there may be definite advantages with respect to noise performance and security when digital is chosen.

Both analog and digital systems can operate at carrier levels which do not cause unusual loading of the cable plant, yet offer noise free reception against white noise sources (such as long amplifier cascades). But no system, no matter how rugged is its design, is totally free from the effects of impulse noise. One advantage of the PCM digital approach is the possibility to interleave (shuffle the order of the samples), so that an impulse noise burst is spread over a longer interval where the error correction coding can compensate without degradation. Also, with digital error detection coding it is possible to take further action if an error can't be corrected. A technique called analog concealment averages samples and substitutes a recent value if an uncorrectable error is detected. While the Dolby ADM digital system works in a different fashion, it also offers the same immunity from noise bursts. By transmitting only small positive or negative increments from the previous value, an error due to a noise burst has a much smaller impact on the signal.

An important aspect of any digital system is the ease of offering a very secure service with little overhead. System operators are increasingly concerned with theft of service, and in general are impressed with the security aspect of digitally based products. Addressability and all its features are easily incorporated into digital audio systems without additional data carriers. For future applications, additional data streams can be multiplexed into the system in a convenient fashion.

Given an uncertain outcome on the long term question of channel efficiency, the attractions of digital audio for consumer acceptance, impulse noise performance, security, addressability and future services provide ample reason to work with the technology today. The cost difference of full featured terminals is not a question since analog and digital products are already on a par. The typical price is in the range of $100-125 depending on quantity.

Service Opportunities

There are three service opportunities which are becoming of interest: 1)enhanced audio, 2) premium audio and 3) pay-per-record. Enhanced audio is the offering of superior audio quality over that which is available today as part of the TV program. This includes the standard network and local TV station feeds as they upgrade for MTS and cable services such as MTV, VH-1, HBO, Cinemax and others who are also upgrading for higher quality. Any of the broadcast systems which offer better performance than standard TV audio can be used for providing this service. Premium audio is interpreted as a pay service with multiple channels of commercial-free programming. Formats such as rock, contemporary, classical, jazz, news, sports, weather and business updates are all candidates for a premium audio service. Pay-per-record is a future possibility, particularly as new digital audio recording equipment is offered in the consumer marketplace. Possibilities to "publish" lesser popular works as well as special events make this an exciting opportunity for the future.

For premium audio and pay-per-record, high quality audio transmission is considered a must. The CATV operator is in a good position, if he selects a digital audio system, to set the standard and establish these businesses.

DIGITAL AUDIO DESCRIPTION

In June 1983, ATC R&D issued a RFP (Request for Proposal) for developing a digital audio product. After analyzing several responses, Toshiba was chosen for the task. Based on experience gained in the development of digital audio as used in DBS receiving equipment for the Japanese market, the DCAT (Digital Cable Audio Terminal) system was designed (reference 6). The following features indicate the capabilities of this system:

Feature	Specification
Tuning range	88-120 MHz
Input range	+5 to -24 dBmv
Required C/N	23 dB
Ultra Hi-fi	16 bit linear PCM
Frequency response	20-20 KHz
Dynamic range	86 dB
Total Harmonic Distortion	0.015%
Channel efficiency	4 channels/ 6 MHz
Super Hi-fi	14/10 bit NIC
Frequency response	20-15 KHz
Dynamic range	76 dB
Total Harmonic Distortion	0.1%
Channel efficiency	8 channels/ 6 MHz
Program number	99 values, 26 tier levels
Security	multi-keyed encryption
Error correction	BCH (63,56) SEC, DED
Headphone jack	front panel volume control, 600 ohm
RF Input	75 ohm, "F" connector
Line output	Phono jacks (L/R) with adjustable outputs (1-3 v RMS)
Power	105-132 AC (60 Hz), 15W
Size	13.5" (w) x 9.5" (d) x 2.25" (h)
Remote control	optional

As described above, the system has two modes of operation: 1) Ultra mode (16 bit PCM) and 2) Super mode (14/10 bit NIC). The Ultra mode provides the same performance as Compact Disc systems. By using 16 bit linear PCM and a sampling rate of 48 KHz no compromise of audio quality is allowed. The Super mode was designed for greater channel efficiency and yet also delivers high quality audio with a S/N of 76 dB. Samples are taken at a 32 KHz rate, with 14 bits reduced to 10 using NIC (Near Instantaneous Companding) techniques. In addition, pre-emphasis is used to further reduce any effects of quantizing noise. The subjective difference between the two modes varies from listener to listener, but tends to be interpreted as a reduction in "presence" or "fullness" of the material. It is very difficult for the average customer to know which mode is being used.

The data is packaged in one msec frames of 2048 bits each. These encrypted frames include sync information, audio information (one Ultra or two Super stereo pairs), addressing data and error correction codes. The resulting 2.048 Mbit data rate is QPSK encoded and occupies a 1.4 MHz RF channel.

Headend System

The headend system supports all encoding requirements for the digital audio system. It is comprised of the following functional elements: 1) A/D conversion, 2) Frame formatting, 3) QPSK modulator and 4) system controller.

The A/D conversion process is a critical function due to the high S/N capabilities of the system. Of particular importance is the input anti-ailiasing filters which must constrain the upper frequency to be either 20 KHz (Ultra) or 15 KHz (Super). To achieve proper rejection of frequencies above these limits and not sacrifice phase response requires careful design. Also included in this function is the pre-emphasis and NIC circuitry.

The frame formatting is a complex task because of all the elements that are put together on a real-time basis. These include audio sample data which must be interleaved, control information (both system wide and specific terminal dependent) which must be provided, then all data must be encrypted along with the error detection and correction coding.

The QPSK modulator and upconverter requires a mixture of digital and RF design. First, the incoming serial data stream at 2.048 Mbits must be formatted into dibits for the quadrature modulation process. After QPSK modulation and filtering at an IF frequency, an upconverter must be used to place the channel in the 88-120 MHz range.

For addressable system control, a system controller computer and software must interface to the frame formatting function in the headend. Also, as in any one-way CATV control system, appropriate interfaces must be made with local operators and the billing/management system.

Terminal Design

The terminal design has many similarities with existing equipment. As in a baseband video converter, there are tuning and demodulation functions. Similar to a Compact Disc player, there are decoding, reformatting and D/A functions. It is the marriage of these two families of requirements that is unique. Additionally, a desire to use existing components and low cost manufacturing techniques completes a picture of the constraints on the design.

The tuning system is standard, covering a range of 32 MHz (88-120 MHz in the initial product). It is microprocessor controlled with 100 KHz channel steps. For demodulation, a QPSK demodulator IC is used from the Japanese DBS system development.

The PCM decoding involves a reversing of the process that was performed at the headend. The de-encryption process must be performed and de-interleaving of the data is required. Any errors are corrected, or detected for analog concealment. The resulting L/R channel information must be D/A converted and carefully filtered to remove the quantization and pre-emphasis effects. Components were used from previous digital audio projects wherever possible. However, a new design of a security IC was required to handle the de-encryption process in a cost-effective manner.

The user interface is primarily through a set of front panel controls. In addition to a 14 position keypad and two digit LED readout, is a volume control and jack for the headphones. An optional remote control has the same features available on the keypad. Basic features include program selection and a memory function for advance selection of an upcoming program. A front panel volume control for the line outputs was not included for cost reasons. However, rear panel adjustment controls are provided separately for the L and R outputs.

FIELD TRIAL PLANS

The first field trial of the DCAT system is scheduled for May 1985 at ATC's Thornton, Colorado system. It will involve an 8 channel headend and up to 100 terminals. A variety of high quality program sources will be used and performance measurements will be made on the system. A market trial of enhanced and premium audio is then planned for fall 1985 as the next step in the development process.

Engineering Trial

In the engineering trial which is scheduled to commence in May several issues will be of major concern. Obviously, the audio performance of the system is the main feature to be studied. However, other concerns are also of a very practical nature: 1) wiring and compatibility issues at the CATV headend, 2) RF signal level measurement procedures and 3) customer premise interconnect configurations.

The audio performance will be measured at selected sites including the headend and locations throughout the cable plant. To help understand performance, a BER (Bit Error Rate) measurement will lead to an estimate of the RF headroom available for the QPSK modulated signal. Analysis of distortion and ingress effects will also be noted.

At the headend, care will be taken to preserve the signal quality available from a number of sources: 1) Compact Disc. player, 2) M/A-COM VideoCipher II audio from HBO and Cinemax, 3) Wegener Dolby audio from MTV and VH-1 and 4) local TV and FM broadcast sources. Any special precautions that are found to be necessary in the wiring layout will be reported.

New techniques must be derived for making practical field measurements of wideband QPSK modulated signals. An investigation of methods that allow usage of current measurement equipment (such as the Wavetek SAM series) will be necessary for field testing.

Perhaps the most uncertain path is the question of various home configuration possibilities. By noting various examples of real world situations, a better understanding of the cable operator's responsibilities will emerge.

As a conclusion to the trial, a summary report will be generated including information relative to the above concerns. Assuming functional success of the DCAT product, the next step is a market trial of high quality audio for the CATV customer.

Market Trial

At the present time, ATC NBD (New Business Development) is in the planning stages of a full scale market trial of enhanced and premium audio which would start in the fall of 1985. In covering the needs of both enhanced audio and premium services it is expected that 12-16 stereo channels will be carried in a high quality format, such as that provided by DCAT. Features which are important to the market trial are addressability, remote control and audio quality.

SUMMARY

Audio source, recording and reproduction equipment has made tremendous strides froward in the last few years. To complete the picture, improvements must also be made in audio broadcasting. Cable systems have an opportunity to set the standard of excellence through the adoption of high quality audio systems.

A digital audio system (DCAT) has been developed which offers the required level of performance and security from theft of service. An engineering field trial will start in May, with a market trial to follow later this year.

References

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